



## **Application Notes for Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya IP Office - Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required to integrate the Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya IP Office. Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya IP Office via SIP. Although not explicitly tested, these Application Notes would also apply to the Talkphone Wide-Area Emergency Broadcast System (WEBS®) Series Devices, which leverage the same electronics and firmware with a similar subset of features (e.g. paging only with no two-way communication) as the VOIP-500 Series and VOIP-600 Series Phones but differ in form factor and packaging.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes describe the configuration steps required to integrate the Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya IP Office. Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya IP Office via SIP. Although not explicitly tested, these Application Notes would also apply to the Talkphone Wide-Area Emergency Broadcast System (WEBS®) Series Devices, which leverage the same electronics and firmware with a similar subset of features (e.g. paging only with no two-way communication) as the VOIP-500 Series and VOIP-600 Series Phones but differ in form factor and packaging.

## 2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations, Avaya SIP and H.323 telephones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer, and conference, from the Avaya IP phones. Additional telephony features, such as call forward and call coverage, were also verified.

The serviceability testing focused on verifying that the Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations come back into service after re-connecting the Ethernet cable or rebooting the IP Call Station.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Talkphone IP Call Station with Avaya IP Office.
- Inbound and outbound calls between Talkphone IP Call Station and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Inbound and outbound calls between the Talkphone IP Call Station and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, transfer, and 3-way conference, initiated from the Avaya IP phone.
- Use of paging, speed-dial buttons, and number lists on the Talkphone IP Call Station.
- Proper system recovery after a restart of the Talkphone IP Call Station and loss of IP connectivity.

## 2.2. Test Results

All test cases passed with the following observation(s):

- Emergency calls cannot be terminated from the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations. The calls can only be disconnected by the destination phone or upon expiration of the Call Conversation Timer. The Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations dial a list of programmed numbers in a round-robin fashion. If the first number in the list does not answer (i.e., Busy, Out of Order, Invalid number), it will call the next number in line and will keep doing so until the destination answers the call or until the 'Call Conversation Timer' expires.
- Dialing Short codes starting with the wildcard \* to activate telephony features is not applicable to Talkaphone IP Call Stations.
- Talkaphone VOIP-500 and VOIP-600 responded “486 Busy Here” to OPTIONs message kept alive from Avaya IP Office during an active call. This did not impact on the active call but is listed here as observation for reference.

## 2.3. Support

For technical support and information on Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations, contact Talkaphone support at:

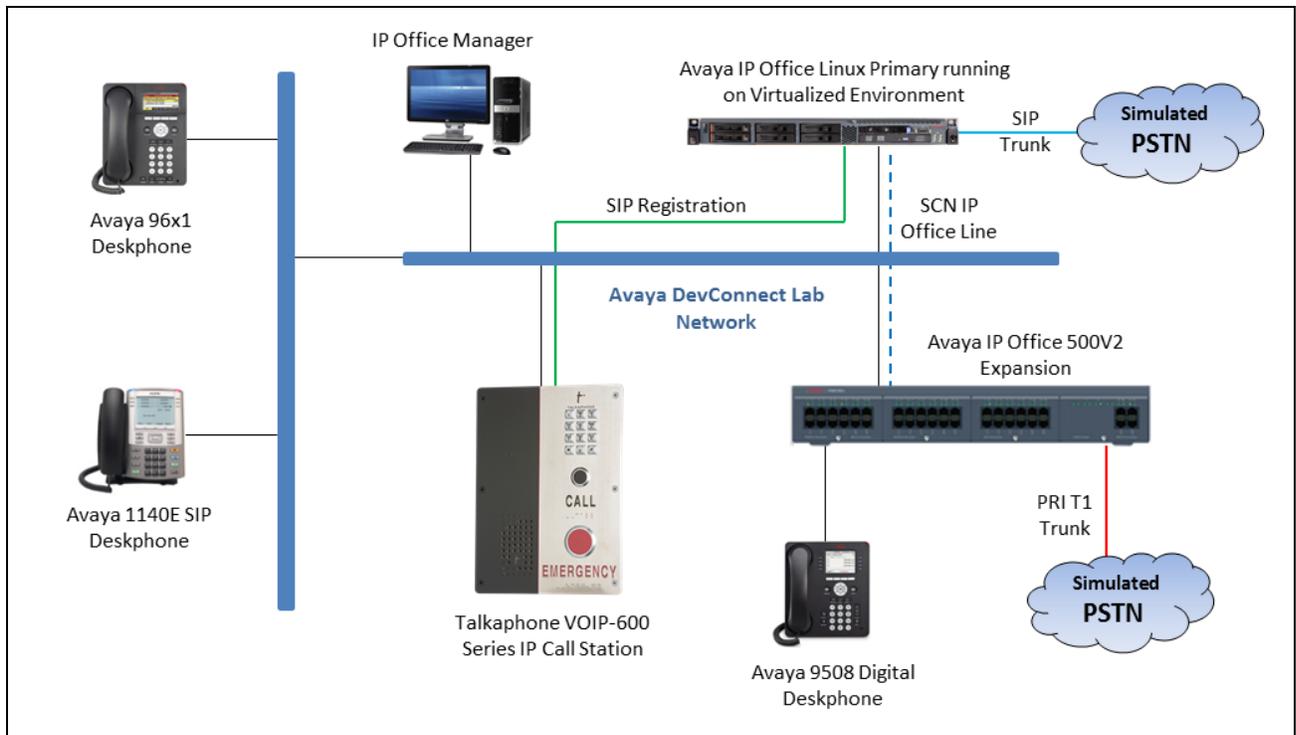
Address : 7530 North Natchez Ave.  
Niles, IL 60714  
Telephone : (773) 539-1100  
Fax : (773) 539-1241  
Email : [info@talkaphone.com](mailto:info@talkaphone.com)  
Web : [www.talkaphone.com](http://www.talkaphone.com)

### 3. Reference Configuration

**Figure 1** illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya IP Office Primary Linux Server Edition running in a virtualized environment with a 500V2 Expansion.
- Avaya IP Office Primary connected to simulated PSTN via SIP trunk.
- Avaya IP Office 500V2 Expansion connected to simulated PSTN via PRI trunk.
- Avaya 96x1 Series H.323 Deskphone and Avaya 1140E SIP Deskphone.
- Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations.

Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya IP Office Primary Linux server.



**Figure 1: Avaya SIP Network with Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations**

## 4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

<b>Equipment/Software</b>	<b>Release/Version</b>
Avaya IP Office Primary Server Edition running on a Virtual Environment	10.0.0.1.0 Build 53
Avaya IP Office 500V2 Expansion	10.0.0.1.0 Build 53
Avaya IP Office Manager running on Microsoft Windows 7	10.0.0.1.0 Build 53
Avaya 96x1 H323 Deskphone	6.629
Avaya 1140E SIP Deskphone	4.0.4.23
Avaya 9508 Digital Deskphone	R45
Talkphone VOIP-500 Series IP Call Stations	Firmware Version : 1.0.2.7j Bootloader Version : 1.1.9
Talkphone VOIP-600 Series IP Call Stations	Firmware Version : 1.0.2.7j Bootloader Version : 1.1.9

## 5. Configure Avaya IP Office

This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify IP Office license.
- Obtain LAN IP address.
- Administer SIP registrar.
- Administer SIP extensions.
- Administer SIP users.
- Administer Internal Twinning.

### 5.1. Verify IP Office License

From a PC running the Avaya IP Office Manager application, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application (not shown). Select the proper IP Office system, and log in using the appropriate credentials.

The **Avaya IP Office Manager** screen is displayed. From the configuration tree in the left pane, select **License**, the list of license displayed in the right panel. Verify that the **3rd Party IP Endpoints** status is “**Valid**”.

The screenshot shows the Avaya IP Office Manager License configuration window. The left pane displays a configuration tree with the following structure:

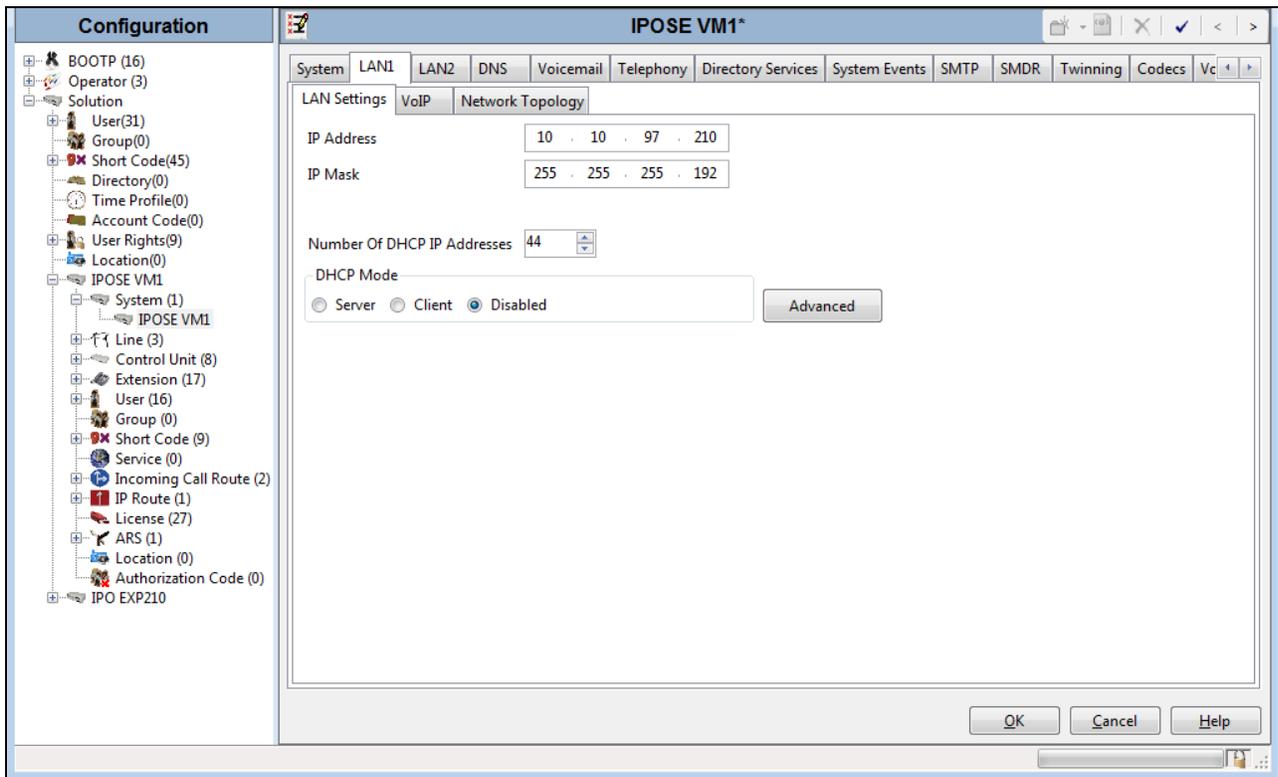
- Configuration
  - BOOTP (16)
  - Operator (3)
  - Solution
    - User (31)
    - Group (0)
    - Short Code (45)
    - Directory (0)
    - Time Profile (0)
    - Account Code (0)
    - User Rights (9)
    - Location (0)
    - IPOSE VMI
      - System (1)
      - Line (3)
      - Control Unit (8)
      - Extension (17)
      - User (16)
      - Group (0)
      - Short Code (9)
      - Service (0)
      - Incoming Call Route
      - IP Route (1)
      - License (27)
      - ARS (1)
      - Location (0)
      - Authorization Code (0)
      - IPO EXP210

The main pane displays the License configuration for a Remote Server. The Licensed Version is 9.1. The System ID (ADI) is 69643e6a3711a282e1b2d6de45c79477b8228770. The PLDS Host ID is 663017273556. The PLDS File Status is Not Present / Invalid.

Feature	License Key	Instances	Status	Expiry Date
VMPro Networked Messaging	i1	W 255	Obsolete	Never
VMPro TTS (Scansoft)	G	2e 255	Obsolete	Never
VMPro TTS (Generic)	V	LB 255	Obsolete	Never
Software Upgrade 255	Z	1	Obsolete	Never
Avaya Softphone License	0	DC 255	Obsolete	Never
CTI Link Pro	y	255	Valid	Never
Wave User	D	cB 255	Valid	Never
Receptionist	A	255	Valid	Never
Preferred Edition Additional Voice...	q	ULB 255	Valid	Never
<b>3rd Party IP Endpoints</b>	<b>A</b>	<b>255</b>	<b>Valid</b>	<b>Never</b>
VMPro Networked Messaging	s	M 255	Obsolete	Never
VMPro Recordings Administrators	q	: 255	Valid	Never
VMPro TTS (Scansoft)	v	DMFe 255	Obsolete	Never
VMPro TTS (Generic)	2	255	Obsolete	Never
SIP Trunk Channels	a	m 255	Valid	Never
Avaya IP endpoints	h	255	Valid	Never

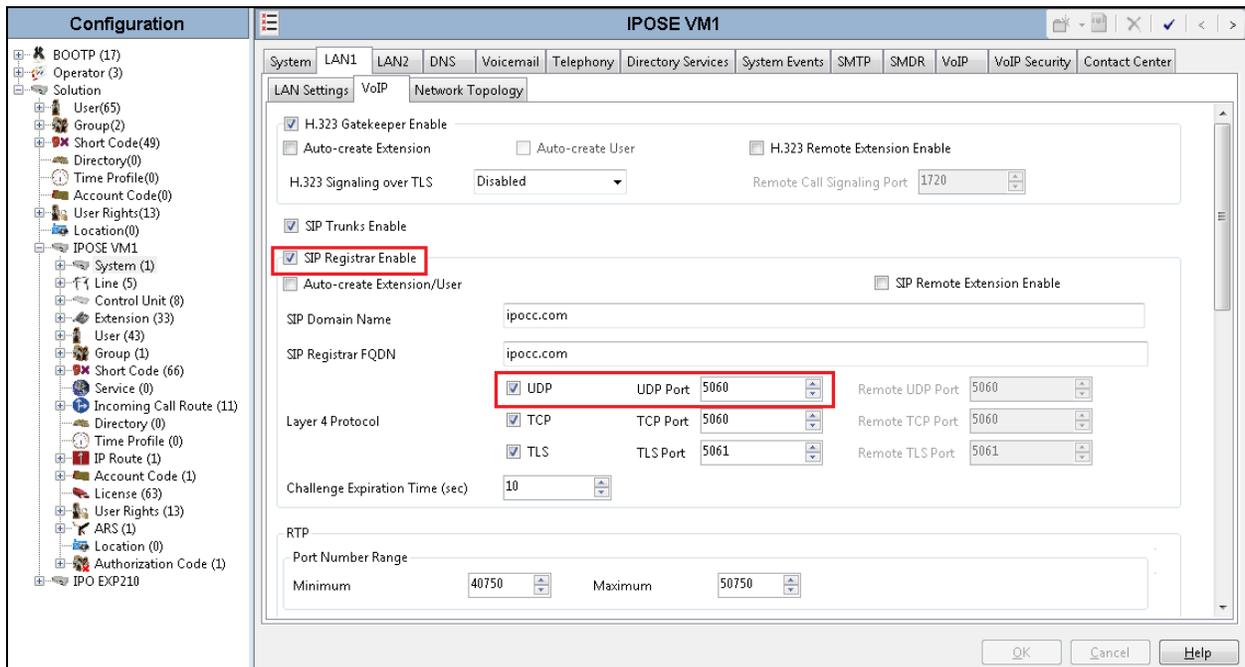
## 5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **IPOSE VM1** screen in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure Talkphone VOIP station. Note that IP Office can support SIP extensions on the **LAN1** and/or **LAN2** interfaces, and the compliance testing used the **LAN1** interface.



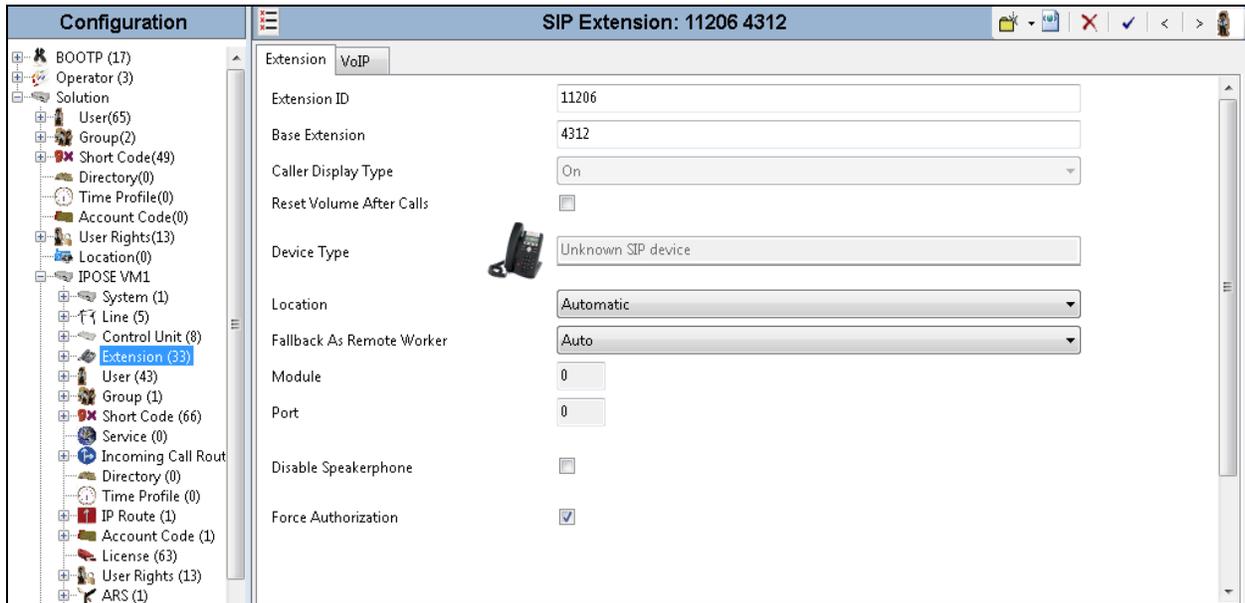
### 5.3. Administer SIP Registrar

Continuing from Section 5.2, select the **VoIP** sub-tab. Make certain that **SIP Registrar Enable** is checked, as shown below. SIP endpoint can use either SIP domain or IP address of LAN1 to register Avaya IP Office, in the compliance a SIP domain was configured in **Domain Name** field but Talkphone VOIP station used IP address of LAN1 to register. Check **UDP checkbox** and enter the default port 5060 in UDP Port since Talkphone VOIP station supports UDP protocol.

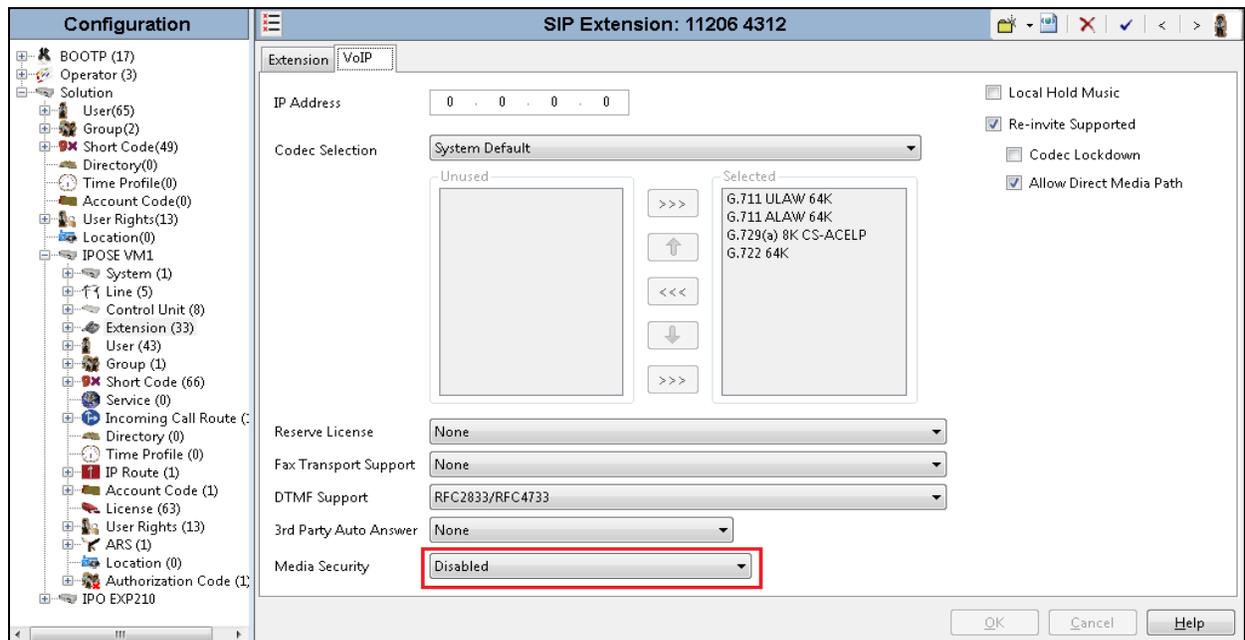


## 5.4. Administer SIP Extensions

From the configuration tree in the left pane, right-click on **Extension**, and select **New** → **SIP Extension** (not shown) from the pop-up list to add a new SIP extension. For **Base Extension**, enter the extension **4312**. Retain the default values in the remaining fields.



Select the **VoIP** tab, select *Disable* in the **Media Security** dropdown menu and retain the default values in all fields. Note that if Media Security is enabled in IP Office System it should be disabled for 3<sup>rd</sup> party endpoint that is not supporting Media Security to avoid audio issue.



## 5.5. Administer SIP User

From the configuration tree in the left pane; right-click on **User** tab and select **New** (not shown) from the pop-up list. Enter desired values for **Name**. For **Extension**, enter the extension from **Section 5.4**. Remember these values as they will be needed to register Talkphone VOIP station to IP Office.

Enter desired values for **Password**, this password is used when user want to login IP Office Softphone.

The screenshot shows the configuration window for a user named 'Talkphone 500' with extension '4312'. The 'User' tab is active, and the configuration fields are as follows:

Name	Talkphone 500
Password	••••••
Confirm Password	••••••
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	Talkphone 500
Extension	4312
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User
	<input type="checkbox"/> Receptionist

Select the **Telephony** tab, followed by the **Supervisor Settings** sub-tab, and enter a desired **Login Code**. This **Login Code** is needed to register Talkphone VOIP station to IP Office.

The screenshot shows the configuration window for a user named 'Talkphone 500' with extension '4312'. The 'Telephony' tab is active, and the 'Supervisor Settings' sub-tab is selected. The configuration fields are as follows:

Login Code	••••••	<input type="checkbox"/> Force Login
Confirm Login Code	••••••	<input type="checkbox"/> Force Account Code
Login Idle Period (sec)		<input type="checkbox"/> Force Authorization Code
Monitor Group	<None>	<input type="checkbox"/> Incoming Call Bar
Coverage Group	<None>	<input type="checkbox"/> Outgoing Call Bar
Status on No-Answer	Logged On (No change)	<input type="checkbox"/> Inhibit Off-Switch Forward/Transfer
IPOCC Agent Type	<None>	<input type="checkbox"/> Can Intrude
Reset Longest Idle Time	<input checked="" type="radio"/> All Calls <input type="radio"/> External Incoming	<input checked="" type="checkbox"/> Cannot Be Intruded
		<input type="checkbox"/> Can Trace Calls
		<input type="checkbox"/> Deny Auto Intercom Calls

## 6. Configure Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations

This section covers the configuration of the Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations. The following procedures are covered:

1. Launching the Web Administration Interface
2. Network Configuration
3. SIP Configuration
4. Configure Audio Settings
5. Configure Call Parameters
6. Configure Buttons

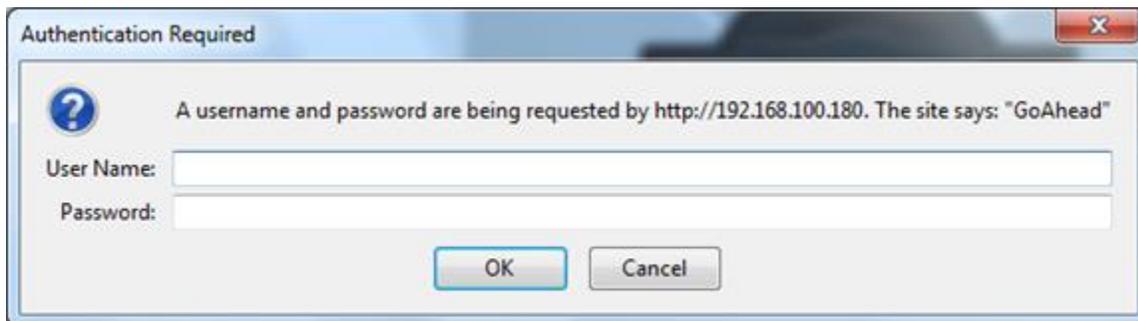
For more information on configuring other features of the Talkphone IP Call Stations, refer to [3].

### 6.1. Launching the Web Administration Interface

The Talkphone IP Call Stations are pre-configured with the following default values:

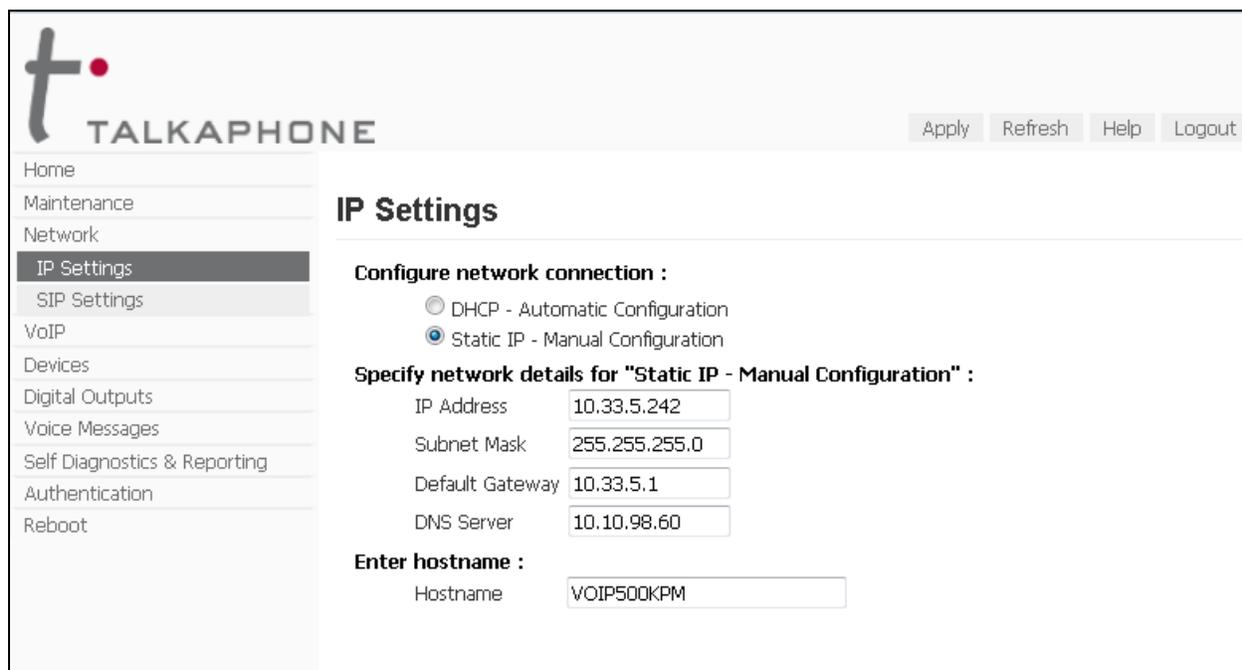
- **IP Address:** 192.168.1.10
- **Username:** admin
- **Password:** admin@123

Ensure that the administration PC and Talkphone VOIP Call Station are connected to the LAN. Open a web browser and enter the IP address of the Talkphone VOIP Call Station in the URL field. The browser prompts for authentication. Log in with the appropriate credentials.



## 6.2. Network Configuration

To modify the IP network configuration of the Talkphone VOIP Call Station, navigate to the **Network** → **IP Settings** page. Configure the IP settings so that it conforms to the customer network requirements. Click **Apply** when done.



The screenshot shows the Talkphone web interface. The top left features the Talkphone logo and a navigation menu with items: Home, Maintenance, Network, IP Settings (highlighted), SIP Settings, VoIP, Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The top right has buttons for Apply, Refresh, Help, and Logout. The main content area is titled "IP Settings" and contains the following configuration options:

- Configure network connection :**
  - DHCP - Automatic Configuration
  - Static IP - Manual Configuration
- Specify network details for "Static IP - Manual Configuration" :**
  - IP Address: 10.33.5.242
  - Subnet Mask: 255.255.255.0
  - Default Gateway: 10.33.5.1
  - DNS Server: 10.10.98.60
- Enter hostname :**
  - Hostname: VOIP500KPM

## 6.3. SIP Configuration

Navigate to **Network** → **SIP Settings** to configure the SIP setting of the Talkphone VOIP Call Station. Configure the following parameters.

Under **Assign a phone number:**

- **Phone Number:** Specify the SIP number (e.g., 4312) configured in **Section 5.4**

Under **Specify SIP Server FQDN/IP Address:**

- **Primary SIP Server FQDN/IP Address:** Specify the IP address of LAN1 of IP Office (e.g., 10.10.97.210) or the SIP domain (e.g., ipocc.com). For the compliance test, the LAN1 IP address was used.

Under **Enable / disable SIP registration:**

- **Register:** Select the checkbox.

Under **Specify SIP registrar** and **Specify outbound proxy**:

- **Username:** Specify the SIP number of the Talkphone IP Call Station (e.g. 4312).
- **Password:** Specify the SIP password configured in **Section 5.5**.
- **Primary SIP Server IP Address:** Specify the LAN IP address of Avaya IP Office (e.g., 10.10.97.210)
- **Port:** Specify the SIP (UDP)port (e.g., 5060).

Accept the default values for the remaining fields and click **Apply** when done.

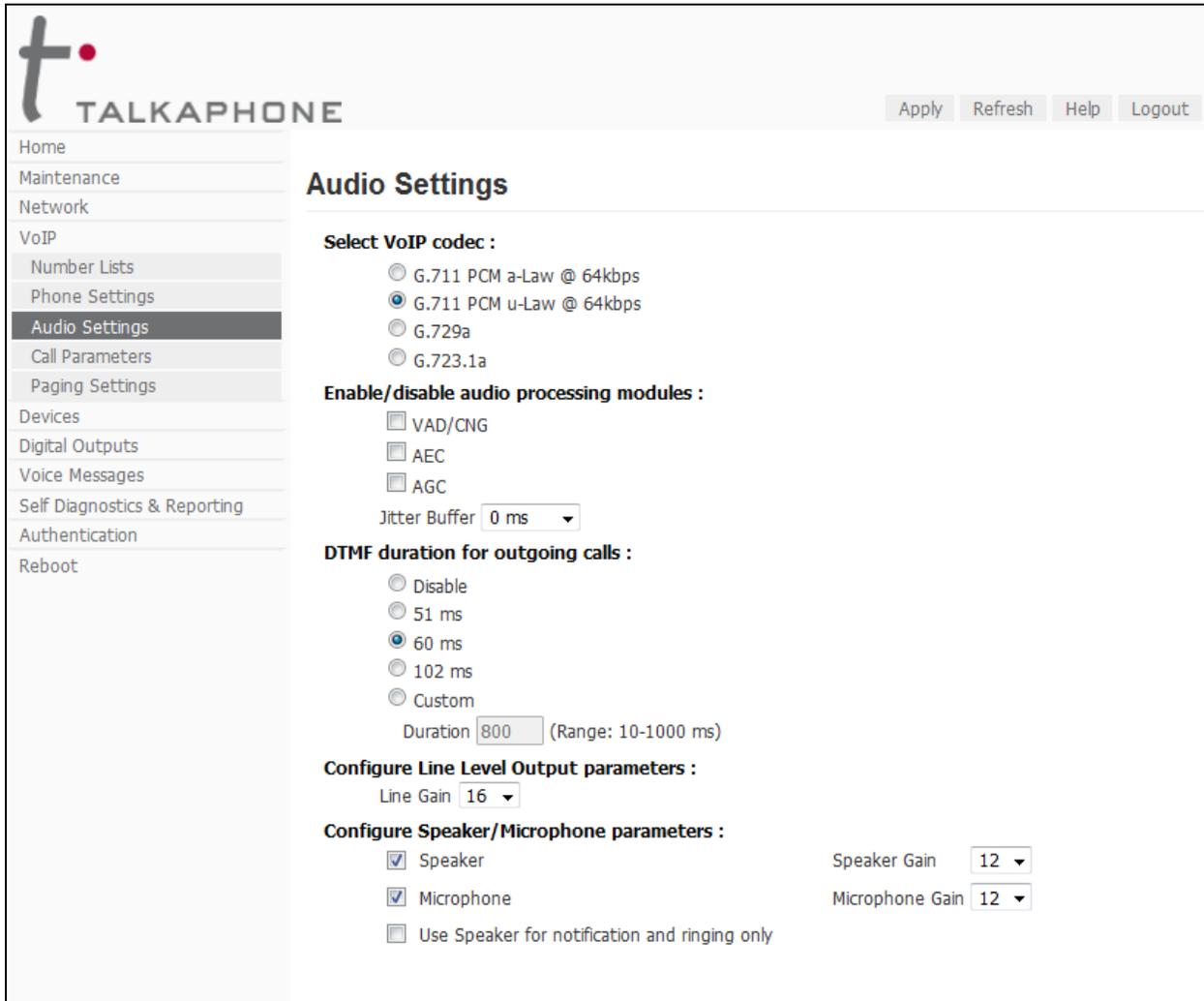
The screenshot displays the 'SIP Settings' page in the Talkphone web interface. The page includes a navigation menu on the left with options like Home, Maintenance, Network, IP Settings, SIP Settings (selected), VoIP, Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The main content area is titled 'SIP Settings' and features a green 'Updated.' status indicator. The settings are organized into several sections:

- Assign a phone number :** Phone Number: 4312
- Specify SIP Server FQDN/IP Address :** Primary SIP Server FQDN/IP Address: 10.10.97.210; Secondary SIP Server FQDN/IP Address: voip.local; Tertiary SIP Server FQDN/IP Address: voip.local
- Enable / disable SIP registration :**  Register
- Specify SIP registrar :** Username: 4312; Password: [masked]; Primary SIP Server IP Address: 10.10.97.210; Secondary SIP Server IP Address: [empty]; Tertiary SIP Server IP Address: [empty]; Port: 5060 (Port Range: 1024-49151); Re-registration Time: 3600 (Range: 10-14400 seconds)
- Specify outbound proxy :** Username: 4312; Password: [masked]; Outbound Proxy 1 IP Address: 10.10.97.210; Outbound Proxy 2 IP Address: [empty]; Outbound Proxy 3 IP Address: [empty]; Port: 5060 (Port Range: 1024-49151)

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## 6.4. Configure Audio Settings

Navigate to **VoIP → Audio Settings** to configure the preferred codec, outbound DTMF duration, and microphone and speaker parameters. All other fields were left at the default values. Click **Apply** when done.



**t TALKAPHONE** Apply Refresh Help Logout

Home  
Maintenance  
Network  
VoIP  
Number Lists  
Phone Settings  
**Audio Settings**  
Call Parameters  
Paging Settings  
Devices  
Digital Outputs  
Voice Messages  
Self Diagnostics & Reporting  
Authentication  
Reboot

### Audio Settings

**Select VoIP codec :**

- G.711 PCM a-Law @ 64kbps
- G.711 PCM u-Law @ 64kbps
- G.729a
- G.723.1a

**Enable/disable audio processing modules :**

- VAD/CNG
- AEC
- AGC

Jitter Buffer

**DTMF duration for outgoing calls :**

- Disable
- 51 ms
- 60 ms
- 102 ms
- Custom

Duration  (Range: 10-1000 ms)

**Configure Line Level Output parameters :**

Line Gain

**Configure Speaker/Microphone parameters :**

- Speaker
- Microphone
- Use Speaker for notification and ringing only

Speaker Gain

Microphone Gain

## 6.5. Configure Call Parameters

Navigate to **VoIP → Call Parameters** to view and customize any of the call parameters, such as **Local Interdigit Timer**, which dictates how long to wait before initiating a call after the user dials the digits, or the **Call conversation Timer**, which specifies how long an emergency call should remain active, unless the far-end drops the call. The following screen shows the default values for the call parameters.

**Note:** After a number is dialed on the Talkphone IP Call Station, the **Local Interdigit Timer** must expire before the call is initiated. The minimum value for the **Local Interdigit Timer** is 5 secs.

The screenshot displays the 'Call Parameters' configuration page in the Talkphone web interface. The page is titled 'Call Parameters' and features a navigation menu on the left with options like Home, Maintenance, Network, VoIP, and Devices. The main content area is divided into several sections:

- Enable/disable call progress tones :** Includes radio buttons for 'Enable' (selected) and 'Disable'.
- Specify key to answer and/or disconnect a call from the Remote Side :** Includes dropdown menus for 'To disconnect a call, press' (set to '# key') and 'To answer a call, press' (set to 'Disable').
- Enable/disable "Welcome Tone" :** Includes radio buttons for 'Enable' (selected) and 'Disable'.
- Configure required timers :** A list of timers with input fields and ranges:
  - Provisional Timer: 5 (Range: 5-20 seconds)
  - Ringer Timer: 5 (Range: 1-12 rings)
  - Hang-up Timer: 0.5 (Range: 0.5-3.0 seconds)
  - Local Interdigit Timer: 5 (Range: 5-20 seconds)
  - Remote Interdigit Timer: 5 (Range: 5-20 seconds)
- Configure optional timers :** A list of optional timers with checkboxes and input fields:
  - Call conversation Timer:  12 (Range: 1-360 min.)
  - Ringback or Busy Timer:  15 (Range: 1-60 seconds)
  - Hang-up On Silence Timer:  30 (Range: 10-360 seconds)

## 6.6. Configure Buttons

Navigate to **Devices** → **Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below.

The screenshot displays the Talkphone web interface. On the left is a navigation menu with the following items: Home, Maintenance, Network, VoIP, Devices, Buttons (highlighted), Keypad, Auxiliary Inputs, LEDs, Auxiliary Outputs, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The main content area is titled "Buttons" and contains two configuration sections:

- Configure Button #1 :**
  - Button #1 Mode: Always Autodial
  - Call from Number List: List 1 (dropdown)
  - Call Priority: 1 (dropdown)
  - Network Priority: 46 (text input) (Range: 0-63)
- Configure Button #2 :**
  - Button #2 Mode: Hook Switch (dropdown)
  - Call from Number List: List 1 (dropdown)
  - Call Priority: 2 (dropdown)
  - Network Priority: 0 (text input) (Range: 0-63)

At the top right of the interface, there are buttons for "Apply", "Refresh", "Help", and "Logout". The Talkphone logo is visible in the top left corner.

## 7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya IP Office.

1. Verify that the Talkphone IP Call Station has successfully registered with Avaya IP Office. In IP Office, use IP Office System Status to check the registration status.

The screenshot displays the Avaya IP Office System Status web interface. The title bar indicates the system is running on IPOSE VM1 (135.10.97.210) on an IP Office Linux PC (10.0.0.1.0 build 53). The main header shows the Avaya logo and the title "IP Office System Status". A navigation menu on the left includes System, Alarms (9), Extensions (7), Trunks (5), Active Calls, Resources, Voicemail, IP Networking, and Locations. The Extensions (7) menu is expanded, showing a list of extensions: 4300, 4301, 4303, 4312 (selected), 4320, 4321, and 4322. The main content area displays the "Extension Status" for extension 4312. The status information includes:

- Extension Number: 4312
- IP address: 10.33.5.242
- Standard Location: None
- Registrar: Primary
- Telephone Type: Unknown SIP Device
- User Agent: ADI VoIP Phone
- Media Stream: RTP
- Layer 4 Protocol: UDP
- Current User Extension Number: 4312
- Current User Name: Talkphone 500
- Forwarding: Off
- Twinning: Off
- Do Not Disturb: Off
- Message Waiting: Off
- Phone Manager Type: None
- SIP Device Features: REFER,UPDATE
- License Reserved: No
- Last Date and Time License Allocated: 1/12/2017 2:42:30 PM
- Packet Loss Fraction: (empty)
- Jitter: (empty)
- Round Trip Delay: (empty)
- Connection Type: (empty)
- Codec: (empty)
- Remote Media Address: (empty)

Below the status information is a table showing call details:

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	01:05:23			

At the bottom of the interface, there are buttons for Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The status bar at the bottom right shows the time as 3:47:54 PM and the system is Online.

Alternatively, the **SIP Settings** screen on the Talkphone IP Call Station also shows the **Registration Status** with the green circle to indicate the registration status successfully.

**TALKPHONE** Apply Refresh Help Logout

Home  
Maintenance  
Network  
IP Settings  
**SIP Settings**  
VoIP  
Devices  
Digital Outputs  
Voice Messages  
Self Diagnostics & Reporting  
Authentication  
Reboot

### SIP Settings

**Assign a phone number :**  
Phone Number: 4312

**Specify SIP Server FQDN/IP Address :**  
Primary SIP Server FQDN/IP Address: 10.10.97.210  
Secondary SIP Server FQDN/IP Address: voip.local  
Tertiary SIP Server FQDN/IP Address: voip.local

**Enable / disable SIP registration :**  
 Register

**Specify SIP registrar :**  
Username: 4312  
Password: ●●●●●●  
Primary SIP Server IP Address: 10.10.97.210  
Secondary SIP Server IP Address:   
Tertiary SIP Server IP Address:   
Port: 5060 (Port Range: 1024-49151)  
Re-registration Time: 3600 (Range: 10-14400 seconds)

**Specify outbound proxy :**  
Username: 4312  
Password: ●●●●●●  
Outbound Proxy 1 IP Address: 10.10.97.210  
Outbound Proxy 2 IP Address:   
Outbound Proxy 3 IP Address:   
Port: 5060 (Port Range: 1024-49151)

**Registration status :**  
 Primary registrar is active : Registered as 4312@135.10.97.210: Pri Reg: 0, Sec Reg: 0, Ter Reg: 0

2. Verify 2-way audio and proper call termination.

## 8. Conclusion

These Application Notes have described the administration steps required to integrate the Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya IP Office Server Edition Solution. Talkphone IP Call Stations successfully registered with Avaya IP Office and basic telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

## 9. Additional References

This section references the Avaya and Talkphone documentation relevant to these Application Notes. The following Avaya product documentation is available at [support.avaya.com](http://support.avaya.com).

Avaya IP Office Documents:

- [1] Administering Avaya IP Office™ Platform with Manager, Release 10, Issue 10.33, October 2016.
- [2] Deploying Avaya IP Office™ Platform Servers as Virtual Machines, Release 10, November 20156.
- [3] IP Office™ Platform 9.1 Using IP Office System Monitor, Release 10, September 2016.
- [4] Administering Avaya IP Office with Manager, Release 10, September 2016.

The following Talkphone documentation may be found at [www.talkphone.com](http://www.talkphone.com).

- [5] *Talkphone VOIP-500 Series Phone Configuration and Operation Manual v3.0.2*, Rev 7/31/2012.
- [6] *Talkphone VOIP-600 Series Configuration and Operation Manual v1.0.1*, Rev 9/17/2014.

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